Troubleshoot busyout Dial-Peers on CUBE or IOS Voice Gateway after IOS Upgrades Are busyout

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Introduction

This document describes the known issue of Cisco Unified Border Elements (CUBE)/voice gateway dial peers status busyout and call failures after Cisco IOS® upgrade to 16.12.6/17.3.5/17.6.1/17.7.1 or higher versions.

Prerequisites

Requirements

There are no specific requirements for this document.

Components Used

The information in this document is based on CUBE.

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, ensure that you understand the potential impact of any command.

Problem

The calls are failures via Cisco IOS Voice Gateway or CUBE after Cisco IOS upgrade to 16.12.6/17.3.5/17.6.1/17.7.1 or higher versions.

Symptoms

When CUBE receives a SIP call and matches an outgoing dial-peer with a 'session server-group' and 'sip options-keepalive' configured, the call fails at the Call Control Application Programming Interface (CCAPI) layer with a 'Cause Value' of 188.

The CUBE doesn't send out outbound INVITE to the destination servers that are part of the server group.

The incoming INVITE is responded with TRYING and 503 Service Unavailable.

The same behavior is observed even when the dial-peer shows as busyout or active KEEPALIVE status under 'show dial-peer voice summary'.

Sample configuration/dial-peer status/debug snippet:

```
dial-peer voice 1000 voip destination-pattern ^1000$ session protocol sipv2 session transport tcp session server-group 1 voice-class sip options-keepalive profile 1 voice-class sip bind control source-interface GigabitEthernet0/0/1 voice-class sip bind media source-interface GigabitEthernet0/0/1 dtmf-relay rtp-nte sip-kpml codec g711ulaw ip qos dscp cs3 signaling no vad voice class server-group 1 ipv4 10.106.117.11 ipv4 10.106.117.6 preference 1
```

```
show dial-peer voice summary AD PRE PASS SESS-SER-GRP OUT TAG TYPE MIN OPER PREFIX DEST-PATTERN FER THRU SESS-TARGET STAT PORT KEEPALIVE VRF 3001 voip up up 0 syst NA 1000 voip up up ^1000$ 0 syst SESS-SVR-GRP: 1 busyout NA show dial-peer voice summary AD PRE PASS SESS-SER-GRP OUT TAG TYPE MIN OPER PREFIX DEST-PATTERN FER THRU SESS-TARGET STAT PORT KEEPALIVE VRF 3001 voip up up 0 syst NA 1000 voip up up ^1000$ 0 syst SESS-SVR-GRP: 1 active NA
```

```
Debug snippet: 007592: Apr 7 07:28:56.046: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg: Received: INVITE sip:1000@10.106.117.5:5060 SIP/2.0 Via: SIP/2.0/UDP 10.106.117.2:5060;branch=z9hG4bK51889 Remote-Party-ID: <sip:3001@10.106.117.2>;party=calling;screen=no;privacy=off From: <sip:3001@10.106.117.2>;tag=12EE76F8-154A To: <sip:1000@10.106.117.5> Date: Wed, 06 Apr 2022 18:28:16 GMT Call-ID: 28E9846D-B50E11EC-8025D5B1-C2D1F237@10.106.117.2 Supported: 100rel,timer,resource-priority,replaces,sdp-anat Min-SE: 1800 Cisco-Guid: 0678152134-3037598188-2149635505-3268538935 User-Agent: Cisco-SIPGateway/IOS-12.x Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER CSeq: 101 INVITE Max-Forwards: 70 Timestamp: 1649269696 Contact: <sip:3001@10.106.117.2:5060> Expires: 180 Allow-Events: telephone-event Content-Type: application/sdp Content-Disposition: session;handling=required Content-Length: 247 v=0 o=CiscoSystemsSIP-GW-UserAgent 8965 7288 IN IP4 10.106.117.2 s=SIP Call c=IN IP4 10.106.117.2 t=0 0 m=audio 18406 RTP/AVP 0 101 c=IN IP4 10.106.117.2 a=rtpmap:0 PCMU/8000 a=rtpmap:101 telephone-event/8000 a=fmtp:101 0-16 a=ptime:20 007649: Apr 7 07:28:56.050://-1/286BC7C68020/SIP/Info/info/2048/sipSPIGetCallConfig: Peer tag 3001 matched for incoming call 007872: Apr 7 07:28:56.061: //89/286BC7C68020/CCAPI/ccCallSetupRequest: Destination=, Calling IE Present=TRUE, Mode=0, Outgoing Dial-peer=1000, Params=0x7FF65E441DE8, Progress Indication=NULL(0) 007935: Apr 7 07:28:56.064: //-1/xxxxxxxxxxxx/SIP/Info/critical/8192/ccsip_call_setup_request: SIP Dialpeer 1000 busied out due to options-keepalive profile in server group 008160: Apr 7 07:28:56.073: //90/286BC7C68020/CCAPI/cc_api_call_disconnected: Cause Value=188, Interface=0x7FF64F4542E8, Call Id=90 008199: Apr 7 07:28:56.077: //89/286BC7C68020/CCAPI/ccCallDisconnect: Cause Value=188, Tag=0x0, Call Entry(Previous Disconnect Cause=0, Disconnect Cause=0) 008239: Apr 7 07:28:56.079: //89/286BC7C68020/SIP/Msg/ccsipDisplayMsg: Sent: SIP/2.0 503 Service Unavailable Via: SIP/2.0/UDP 10.106.117.2:5060;branch=z9hG4bK51889 From: <sip:3001@10.106.117.2>;tag=12EE76F8-154A To: <sip:1000@10.106.117.5>;tag=1C2F76-17F5 Date: Wed, 06 Apr 2022 17:28:56 GMT Call-ID: 28E9846D-B50E11EC-8025D5B1-C2D1F237@10.106.117.2 Timestamp: 1649269696 CSeq: 101 INVITE Allow-Events: telephone-event Server: Cisco-SIPGateway/IOS-17.3.5 Reason: Q.850;cause=0 Session-ID: 00000000000000000000000000000000;remote=3c1f754eba075201a684fda2c51c04df Content-Length: 0
```

Workaround

1. Configure the outgoing dial-peer with 'session target ip4:', instead of 'session server-group'. if needed, create a separate dial-peer for each IP of the server group.

```
dial-peer voice 1000 voip session target ipv4:x.x.x.x dial-peer voice 1001 voip session target ipv4:x.x.x.x
```

2. Remove the 'sip options-keepalive' on the dial-peer.

```
dial-peer voice 1000 voip no voice-class sip options-keepalive profile 1
```

3. Downgrade to an earlier version, this issue was introduced after the commit of Cisco bug ID CSCvx92872.

This issue is documented by Cisco bug ID CSCvz80171.